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TITLE: ECHO CANCELER

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INT-CL (IPC): H04B003/23, H03H021/00 , H04M001/00 , H04M001/60

ABSTRACT:

PURPOSE: To cancel echo with pseudo echo generated based on the delay time of a returned reference signal by sending the reference signal defining the specified frequency component of a transmitted signal as a carrier wave.

CONSTITUTION: A ring buffer 12 samples a transmitted signal T in the cycle of 125μs and time sequentially stores amplitude data for 250ms. A reference signal generator 2 extracts the frequency component of 2400±50Hz in the transmitted signal and sends out the reference signal modulated by the rectangular wave of 250ms as the carrier wave. A level comparator 5 finds an amplitude ratio from the amplitudes of the reference signal returned through a

BPF 7 and the transmitted reference signal. A phase comparator 10 finds the delay time of the transmitted reference signal and the returned reference signal. An echo quantity estimator 11 specifies correspondent data in the ring buffer 12 and outputs the pseudo echo, for which these data are adjusted by the amplitude ratio, through an amplifier 13. An adder 14 subtracts the pseudo echo from a received audio signal and cancels the received echo.

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審査請求　未請求　請求項の数１　ＦＤ　（全　７　頁）

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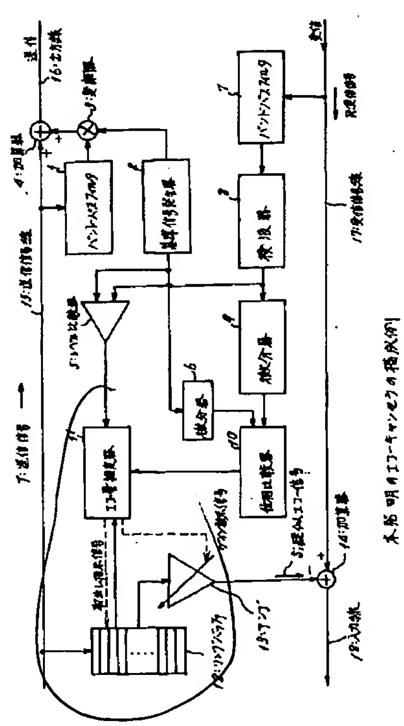
(54)【発明の名称】　エコーキャンセラ

(57)【要約】

【目的】　エコー時間の長短に関わらずキャンセル処理量が一定になるエコーキャンセラを提供すること。

【構成】　自局から送信される音声信号を常時サンプリングしサイクリックに順次時系列に格納するリングバッファ１２と、相手局へ送信する音声信号の特定の周波数成分を抽出し搬送波として使用し周期的に発生する基準信号を送出する基準信号発生２と、該送出する基準信号と相手局から戻って来る基準信号の振幅を比較し振幅比を求めるレベル比較手段５と、該送出する基準信号と相手局から戻って来る基準信号の遅延時間を求める位相比較器１０と、該遅延時間からリングバッファ１２との所定のデータ位置を求め該データを振幅比に合わせて調整し疑似エコーＳを出力するエコー量推定器１１とアンプ１３を具備する疑似エコー出力手段と、相手局から受信した音声信号から該疑似エコーＳを減算し該受信した音声信号に含まれるエコーを打ち消す手段１４を設けたエコーキャンセラ。



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【特許請求の範囲】

【請求項1】　電話機に設けられ、相手局からの音声信号に含まれるエコー信号を打ち消すエコーキャンセラであって、

自局から送信される音声信号を常時サンプリングし、サイクリックに順次時系列に格納する音声データ格納手段と、前記相手局へ送信する音声信号の特定の周波数成分を抽出し搬送波として使用し周期的に発生する基準信号を送出する基準信号送出手段と、該送出する基準信号と前記相手局から戻って来る基準信号の振幅を比較し振幅比を求める振幅比較手段と、該送出する基準信号と前記相手局から戻って来る基準信号の遅延時間を求める位相比較手段と、該遅延時間から前記音声データ格納手段の所定のデータ位置を求め該データを前記振幅比に合わせて調整し疑似エコーを出力する疑似エコー出力手段と、前記相手局から受信した音声信号から該疑似エコーを減算し該受信した音声信号に含まれるエコーを打ち消す手段を設けたことを特徴とするエコーキャンセラ。

【発明の詳細な説明】

【0001】

【産業上の利用分野】本発明は電話回線を介して相手局から戻ってくるエコーを打ち消すエコーキャンセラに関するものである。

【0002】

【従来技術】長距離電話回線では、相手側の電話装置を通して自分の声が遅延されて戻って来るため、エコーとなって通話に著しい障害を与えることがある。これを防ぐための装置がエコーキャンセラであり、電話装置の主装置の送受信回路等に設置される。図3は従来のエコーキャンセラの構成を示すブロック図である。図示するように、従来のエコーキャンセラはＦＩＲフィルタ（非巡回形フィルタ）３０、係数更新部３４、加算器３５を具備する。ＦＩＲフィルタ（非巡回形フィルタ）３０は複数個の遅延素子３１－１、３１－２、・・・、アンプ３２－１、３２－２、・・・加算器３３を有し、係数更新部３４でＦＩＲフィルタ３０の係数（ゲイン）を設定するように構成されている。

【０００３】ＦＩＲフィルタ３０の各遅延素子３１－１、３１－２、・・・は遅延時間が設定されて送信信号線３６から直列に接続され、各接続点からはアンプ３２－１、３２－２、・・・が接続され、各アンプの出力は加算器３３の入力に接続されている。更に、加算器３３の出力は受信信号線３７と共に加算器３５の入力に接続される。加算器３５の出力は係数更新部３４の入力に接続され、係数更新部３４の出力は各アンプのゲインを調整するように接続されている。

【０００４】送信信号は送信信号線３６から局へ送信されると共に、各遅延素子３１－１、３１－２、・・・を通して各アンプに入力され、各アンプの出力信号は加算器３３で加算され疑似エコー信号ｓを発生させる。受信信号線３７で受信した受信信号は加算器３５で疑似エコー信号ｓが減算された後、受信処理されると共に、係数更新部３４へ入力される。係数更新部３４はＣＰＵ（中央処理装置）を具備し、受信信号に含まれるエコー信号が最小になるように各アンプのゲインを調整し、加算器３３は最適な疑似エコー信号ｓを出力する。受信信号に含まれるエコー信号は電話回線により異なり、係数更新部３４は電話回線毎に繰返し学習することにより疑似エコー信号ｓの最適化を図っている。

【０００５】

【発明が解決しようとする課題】しかしながら、上記に述べた従来のエコーキャンセラはエコー時間（送信信号のエコーが戻って来るまでの時間）が長い場合、ＦＩＲフィルタ（非巡回型フィルタ）３０の遅延素子及びアンプの数を増加する必要があり、各アンプの係数（ゲイン）を最適化する処理が膨大な量となり係数更新部３４で処理ができなくなると云う問題があった。

【０００６】本発明は上述の点に鑑みてなされたもので、上記問題点を除去し、エコー時間の長短に関わらずキャンセル処理量が一定になるエコーキャンセラを提供することを目的とする。

【０００７】

【課題を解決するための手段】上記課題を解決するため本発明は、電話機に設けられ、相手局からの音声信号に含まれるエコー信号を打ち消すエコーキャンセラであって、自局から送信される音声信号を常時サンプリングし、サイクリックに順次時系列に格納する音声データ格納手段（１２）と、相手局へ送信する音声信号の特定の周波数成分を抽出し搬送波として使用し周期的に発生する基準信号を送出する基準信号送出手段（２）と、該送出する基準信号と相手局から戻って来る基準信号の振幅を比較し振幅比を求める振幅比較手段（５）と、該送出する基準信号と相手局から戻って来る基準信号の遅延時間を求める位相比較手段（１０）と、該遅延時間から音声データ格納手段（１２）の所定のデータ位置を求め該データを振幅比に合わせて調整し、疑似エコーＳを出力する疑似エコー出力手段（エコー量推定器１１とアンプ１３）と、相手局から受信した音声信号から該疑似エコーＳを減算し、該受信した音声信号に含まれるエコーを打ち消す手段（１４）を設けたことを特徴とする。

【０００８】

【作用】本発明では、上記に説明したように、基準信号の送信時と受信時の振幅比と、基準信号の送信から受信までの遅延時間によりエコーを推定し、格納している音声信号のデータから疑似エコーＳを算出し、受信した音声信号から減算するので、疑似エコー出力手段の処理量はエコー時間の長短に関係なく一定となる。従って、従来のようにハードウエア素子を増やすことなく、エコー時間の長短に関係なく安定した通話が期待できる。また、基準信号を送る搬送波は送信信号から抽出するので

雑音等の障害も少なくなる。

【０００９】

【実施例】以下、本発明の一実施例を図面に基づいて詳細に説明する。図１は本発明のエコーキャンセラの構成を示すブロック図である。図示するように本発明のエコーキャンセラはバンドパスフィルタ１、基準信号発生器２、変調器３、加算器４、レベル比較部５、微分器６、バンドパスフィルタ７、検波器８、微分器９、位相比較器１０、エコー量推定器１１、リングバッファ１２、アンプ１３、加算器１４を具備する。

【００１０】送信信号線１５はバンドパスフィルタ１の入力及び、加算器４の一方の入力に接続され、バンドパスフィルタ１の出力は基準信号発生器（方形波発生器）２の出力と共に変調器３の入力に接続され、変調器３の出力は加算器４の一方の入力に接続され、加算器４の出力は出力線１６に接続されている。

【００１１】受信信号線１７は加算器１４の入力に接続すると共にバンドパスフィルタ７を通して検波器８に接続する。検波器８の出力及び基準信号発生器２の出力はレベル比較器５の入力に接続されると共に、それぞれ微分器９及び微分器６を通して位相比較器１０の入力に接続される。

【００１２】レベル比較器５の出力及び、位相比較器１０の出力はエコー量推定器１１の入力に接続され、その出力はリングバッファ１２及びアンプ１３のゲイン調整端子に接続される。送信信号線１５はリングバッファ１２の入力に接続され、リングバッファ１２の出力はアンプ１３の入力に接続され、アンプ１３の出力は受信信号線１７と共に加算器１４の入力に接続され、該加算器１４の出力は入力線１８に接続されている。

【００１３】リングバッファ１２は送信信号Ｔ（音声信号）を１２５μｓのサンプリング周期で常時サンプリングし、２５０ｍｓ分のデータ（振幅）をサイクリックに順次時系列で格納する記憶装置である。

【００１４】次に動作を説明する。送信信号Ｔ（音声信号）はバンドパスフィルタ１で２．４ＫＨｚ±５０Ｈｚの信号が抽出され、その出力信号は変調器３で基準信号発生器２の出力信号（２５０ｍｓ周期の方形波）で変調（振幅変調）され、変調器３の出力信号は送信信号Ｔと加算器４で加算され出力線１６から出力される。

【００１５】受信信号Ｒは受信信号線１７で受信され、バンドパスフィルタ７を通して２．４ＫＨｚ±５０Ｈｚの信号が抽出されて検波器８で基準信号が復調され、信号振幅が基準信号発生器２の出力信号（２５０ｍｓ周期の方形波）とレベル比較器５で比較され、基準信号の送信時と受信時の振幅比が出力される。更に、検波器８で復調された信号は微分器９で微分され、基準信号発生器２の出力信号は微分器６で微分され、共に立上り部分を抽出し位相比較器１０で位相が比較され、基準信号の送信から受信までの遅延時間がエコー量推定器１１へ出力される。

【００１６】エコー量推定器１１はレベル比較器５の出力信号（振幅比）及び、位相比較器１０の出力信号（遅延時間）からエコー量を推定し、疑似エコー信号を発生させる（詳細後述）。即ち、リングバッファ１２に格納されたデータ（時系列で格納されるデータ）から所定のデータを取り出す位置を指定し、前記データを増幅するアンプ１３のゲインを指示する。アンプ１３はエコー量推定器１１の指示に従ってゲインを設定し前記データに応じた信号（疑似エコー信号Ｓ）を加算器１４へ出力する。加算器１４は受信信号Ｒから疑似エコー信号Ｓを減算し入力線１８へ出力する。

【００１７】図２はエコー量推定器の動作を示すフローチャートである。エコー量推定器１１はＣＰＵ及びメモリ（図では省略）を具備し、本フローチャートを実行するプログラムはエコー量推定器１１のメモリに格納されＣＰＵで実行される。図２のフローチャートに従って説明する。エコー量推定器１１はレベル比較器５及び位相比較器１０の出力信号を読み込む（ステップＳＴ１）。

【００１８】位相比較器１０の出力信号を基にリングバッファ１２の取り出し位置を決める（ステップＳＴ２）。レベル比較器５の出力信号を基にアンプ１３のゲインを決定する（ステップＳＴ３）。リングバッファ１２へデータの出力（アンプ１３への出力）を指示し（ステップＳＴ４）、アンプ１３へゲインの設定を指示し、データを疑似エコー信号Ｓとして出力させる（ステップＳＴ５）。以上のステップＳＴ１～ステップＳＴ５を繰返し実行する。

【００１９】上記説明は発呼側のエコーキャンセンラであり、バンドパスフィルタ１は送信信号Ｔ（音声信号）中で知覚にあまり影響しない２．４ＫＨｚ±５０Ｈｚの狭帯域の信号を抽出し基準信号（２５０ｍｓ周期の方形波）で変調をかけて利用した。また、着呼側のエコーキャンセンラの周波数帯域は発呼側との混信を防ぐために２．６ＫＨｚ±５０Ｈｚを使用すればよい。上記基準信号を乗せる周波数及び周波数帯域、基準信号の周期（２５０ｍｓ）とそれに関連したリングバッファサイズ（２５０ｍｓの格納分）は一例として使用した値であって、他の値を使用してもよい。

【００２０】上記説明したように本実施例によれば、基準信号の送信時と受信時の振幅比と基準信号の送信から受信までの遅延時間によりエコー量を推定し、格納している音声信号のデータから疑似エコー信号Ｓを算出し受信信号Ｒから減算するので、エコー時間に関係なくエコー量推定器１１の処理量は一定となり、電話回線によるエコー時間の長短に関係なく安定した通話が期待できる。また、基準信号を送る搬送波は送信信号から抽出するので雑音等の障害も少なくなる。

【００２１】

【発明の効果】以上説明したように本発明によれば、下

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記のような優れた効果が期待される。基準信号の送信時と受信時の振幅比と、基準信号の送信から受信までの遅延時間によりエコーを推定し、格納している音声信号のデータから疑似エコーSを算出し、受信した音声信号から減算するので、疑似エコー出力手段の処理量はエコー時間の長短に関係なく一定となる。従って、従来のようにハードウエア素子を増やすことなく、エコー時間の長短に関係なく安定した通話が期待できる。また、基準信号を送る搬送波は送信信号から抽出するので雑音等の障害も少なくなる。

【図面の簡単な説明】

【図１】本発明のエコーキャンセラの構成例を示すブロック図である。

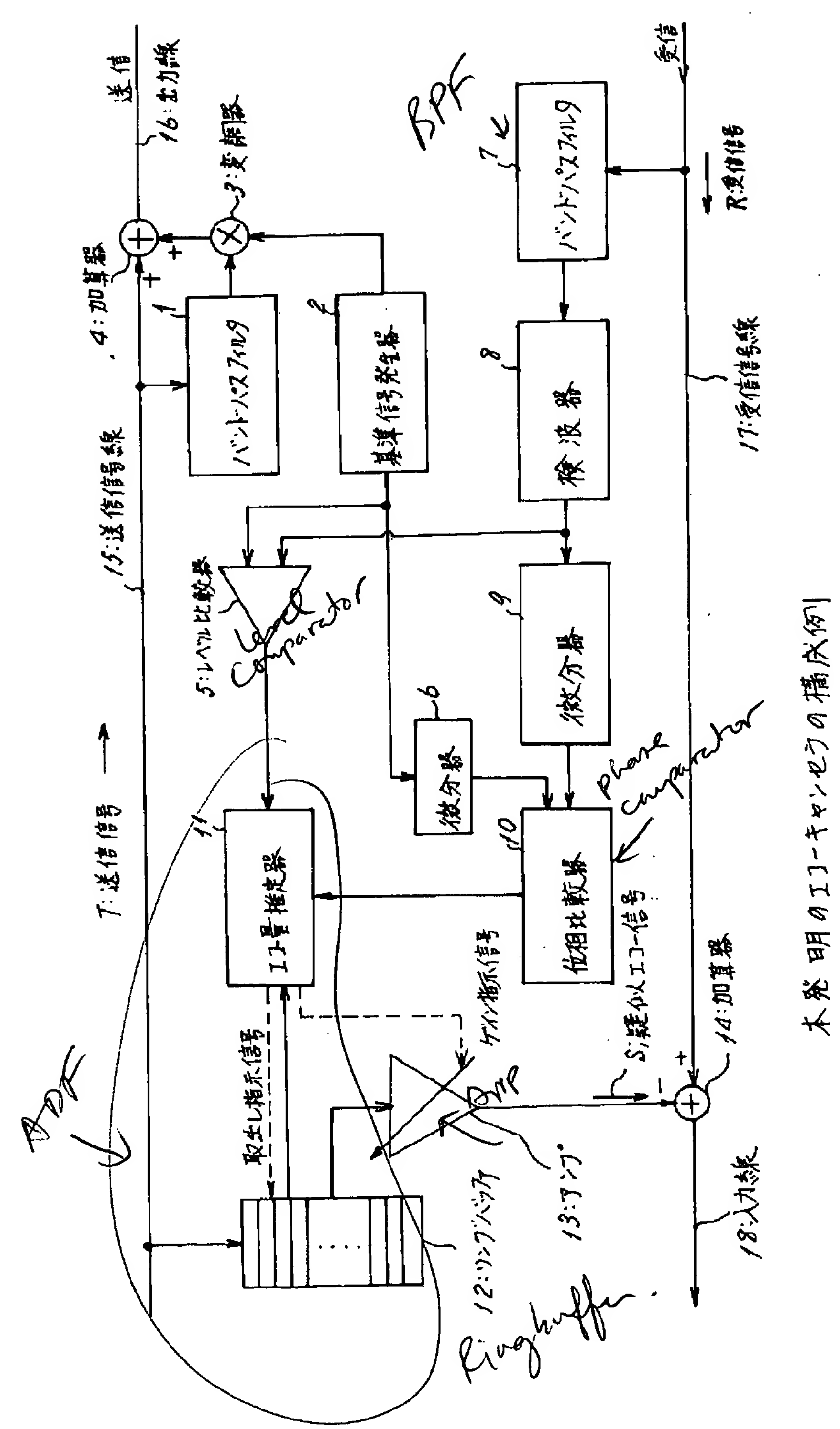
【図２】エコー量推定器の動作を示すフローチャートである。

【図３】従来のエコーキャンセラの構成例を示すブロック図である。

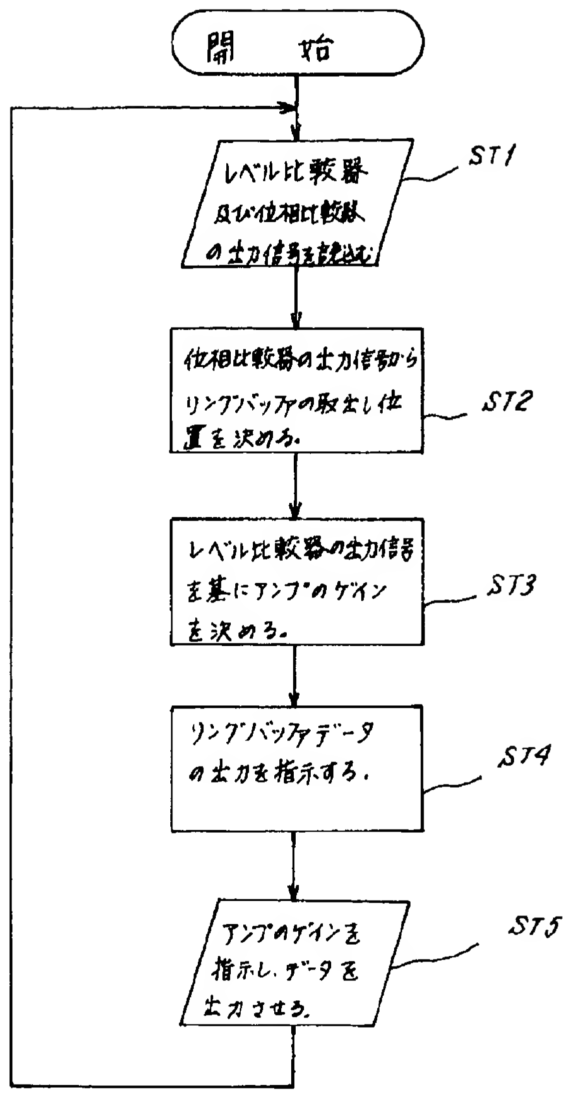
【符号の説明】

1	バンドパスフィルタ
2	基準信号発生器
3	変調器
4	加算器
5	レベル比較部
6	微分器
7	バンドパスフィルタ
8	検波器
9	微分器
10	位相比較器
11	エコー量推定器
12	リングバッファ
13	アンプ
14	加算器

【図1】

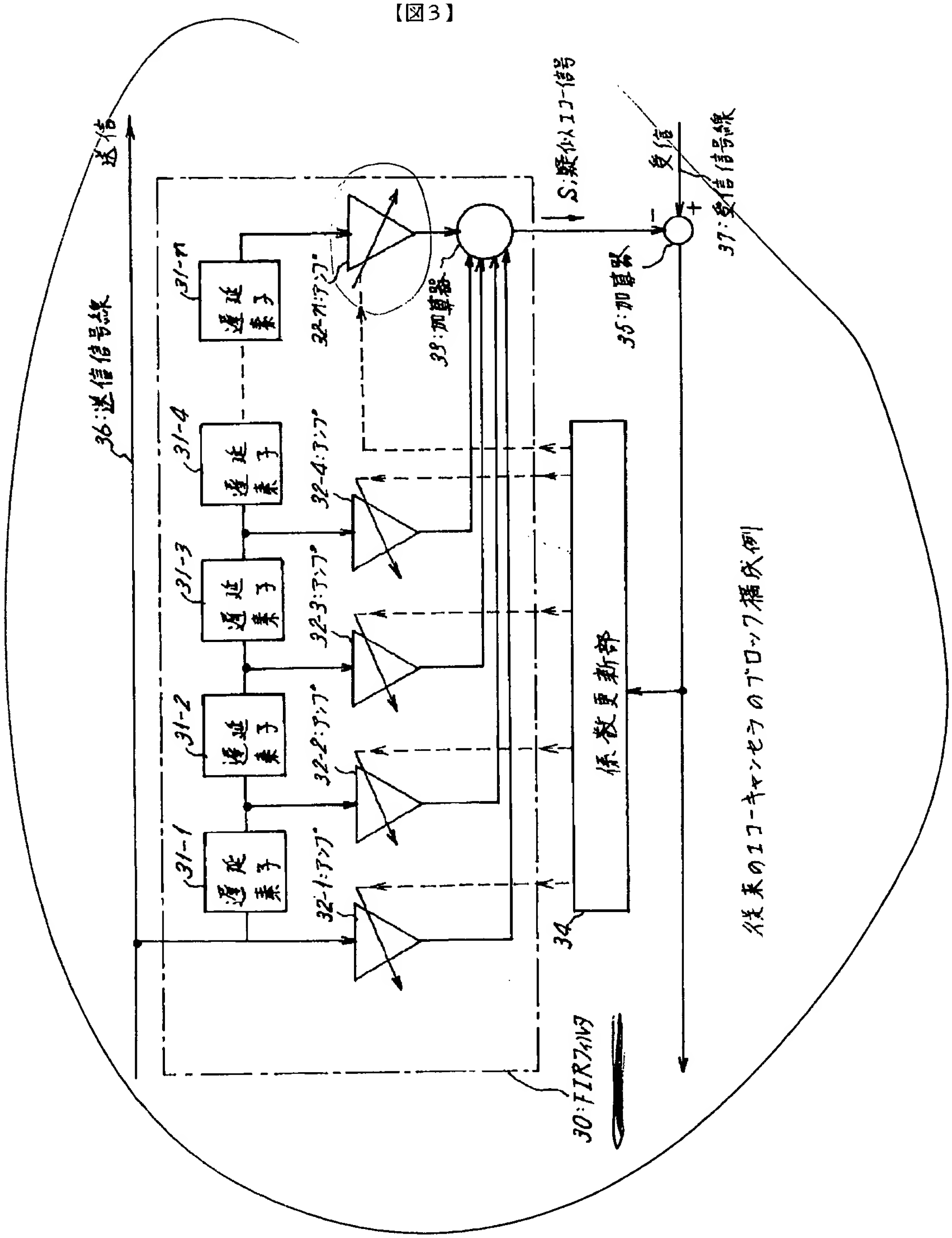

送信
16:出力線
3:変調器
4:加算器
1
バンドパスフィルタ
2
基準信号発生器
BPF
7
バンドパスフィルタ
受信
R:受信信号
8
検波器
17:受信信号線
15:送信信号線
5:レベル比較器
Level Comparator
9
微分器
6
微分器
phase comparator
10
位相比較器
11
エコー量推定器
T:送信信号
ゲイン指示信号
取出し指示信号
S:疑似エコー信号
14:加算器
Amp
13:アンプ
12:リングバッファ
Ringbuffer.
18:入力線
BPF
本発明のエコーキャンセラの構成例

【図2】



エコー量推定器の動作を示すフローチャート

【図3】



送信
36:送信信号線
31-1
31-2
31-3
31-4
31-n
遅延素子
32-1:タップ
32-2:タップ
32-3:タップ
32-4:タップ
32-n:タップ
33:加算器
S:疑似エコー信号
受信
37:受信信号線
35:加算器
係数更新部
34
30:FIRフィルタ
従来のエコーキャンセラのブロック構成例

* NOTICES *

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1.This document has been translated by computer. So the translation may not reflect the original precisely.
2.**** shows the word which can not be translated.
3.In the drawings, any words are not translated.

CLAIMS

[Claim(s)]
[Claim 1] An echo canceller which negates an echo signal which is characterized by providing the following, and which is formed in a telephone set and included in a sound signal from a distant office A voice data storing means to always sample a sound signal transmitted from a local station, and to store in time series one by one cyclically A reference signal sending-out means to send out a reference signal which extracts a specific frequency component of a sound signal which transmits to said distant office, uses as a subcarrier, and is generated periodically A magnitude-comparison means to compare amplitude of a this reference signal to send out and a reference signal which returns from said distant office, and to ask for a gain A means negate the echo included in the sound signal which subtracted this false echo and this received from a phase-comparison means find a time delay of a this reference signal to send out and a reference signal which returns from said distant office, a false echo output means adjusts this data according to said gain in quest of a predetermined data location of said voice-data storing means from this time delay, and output a false echo, and a sound signal received from said distant office

DETAILED DESCRIPTION

[Detailed Description of the Invention]
[0001]
[Industrial Application] This invention relates to the echo canceller which negates the echo which returns from a distant office through the telephone line.
[0002]
[Description of the Prior Art] By the long-distance telephone circuit, since its voice is delayed through the telephone equipment of the other party and it returns, it may become an echo and a remarkable failure may be done to a message. The equipment for preventing this is an echo canceller, and is installed in the transceiver circuit of the main unit of telephone equipment etc. Drawing 3 is the block diagram showing the configuration of the conventional echo canceller. The conventional echo canceller possesses the FIR filter (non-recursive filter) 30, the renewal section 34 of a coefficient, and an adder 35 so that it may illustrate. the delay element 31-1 of plurality [filter / 30 / (non-recursive filter) / FIR], 31-2, ..., amplifier 32-1, 32-2, and ... it has an adder 33, and it is constituted so that the coefficient (gain) of the FIR filter 30 may be set up in the renewal section 34 of a coefficient.
[0003] As for each delay element 31-1 of the FIR filter 30, 31-2, and ..., a time delay is set up, it connects with a serial from the sending-signal line 36, amplifier 32-1, 32-2, and ... are connected from each node, and the output of each amplifier is connected to the input of an adder 33. Furthermore, the output of an adder 33 is connected to the input of an adder 35 with the input-signal line 37. The output of an adder 35 is connected to the input of the renewal section 34 of a coefficient, and the output of the renewal section 34 of a coefficient is connected so that the gain of each amplifier may be adjusted.
[0004] While a sending signal is transmitted to a station from the sending-signal line 36, it is inputted into each amplifier through each delay element 31-1, 31-2, and ..., and the output signal of each amplifier is added with an adder 33, and generates false echo signal s. It is inputted into the renewal section 34 of a coefficient while reception of the input signal which received by the input-signal line 37 is carried out, after false echo signal s is subtracted with an adder 35. The renewal section 34 of a coefficient possesses CPU (central processing unit), it adjusts the gain of each amplifier so that the echo signal included in an input signal may become min, and it outputs false echo signal s with the optimal adder 33. The echo signal included in an input signal changes with telephone lines, and optimization of false echo signal s is in drawing by learning the renewal section 34 of a coefficient repeatedly for every telephone line.
[0005]
[Problem(s) to be Solved by the Invention] However, when echo time (time amount until the echo of a sending signal returns) was long, the conventional echo canceller described above needed to increase the delay element of the FIR filter (non-recursive filter) 30, and the number of amplifier, and the processing which optimizes the coefficient (gain) of each amplifier became a huge amount, and it had the problem which says that processing becomes impossible in the renewal section 34 of a coefficient.
[0006] It aims at offering the echo canceller to which this invention was made in view of the above-mentioned point, removes the above-mentioned trouble, and is not concerned with the merits and demerits of echo time, but cancellation throughput becomes fixed.
[0007]
[Means for Solving the Problem] It is the echo canceller which negates an echo signal which this invention is prepared in a telephone set in order to solve the above-mentioned technical problem, and is included in a sound signal from a distant office. A voice data storing means to always sample a sound signal transmitted from a local station, and to store in time series one by one cyclically (12), A reference signal sending-out means to send out a reference signal which extracts a specific frequency component of a sound signal which transmits to a distant office, uses as a subcarrier, and is generated periodically (2), A magnitude-comparison means to compare amplitude of a this reference signal to send out and a reference signal which returns from a distant office, and to ask for a gain (5), A phase-comparison

means to find a time delay of a this reference signal to send out and a reference signal which returns from a distant office (10), A false echo output means to adjust this data according to a gain in quest of a predetermined data location of a voice data storing means (12) from this time delay, and to output the false echo S (the amount presumption machine 11 of echoes, and amplifier 13), It is characterized by establishing a means (14) to negate an echo which subtracts this false echo S from a sound signal received from a distant office, and is included in a this received sound signal.

[0008]

[Function] In this invention, since an echo is presumed by the time delay from the gain at the time of transmission of a reference signal, and reception, and transmission of a reference signal to reception, and the false echo S is computed from the data of the stored sound signal and it subtracts from the received sound signal as explained above, the throughput of a false echo output means becomes fixed regardless of the merits and demerits of echo time. Therefore, the message stabilized regardless of the merits and demerits of echo time can be expected, without increasing a hardware element like before. Moreover, since the subcarrier which sends a reference signal is extracted from a sending signal, its failures, such as a noise, also decrease.

[0009]

[Example] Hereafter, one example of this invention is explained to details based on a drawing. Drawing 1 is the block diagram showing the configuration of the echo canceller of this invention. The echo canceller of this invention possesses a band pass filter 1, the reference signal generator 2, a modulator 3, an adder 4, the level comparator 5, a differentiator 6, a band pass filter 7, a wave detector 8, a differentiator 9, a phase comparator 10, the amount presumption machine 11 of echoes, the ring buffer 12, amplifier 13, and an adder 14 so that it may illustrate.

[0010] The sending-signal line 15 is connected to the input of a band pass filter 1, and one input of an adder 4, the output of a band pass filter 1 is connected to the input of a modulator 3 with the output of the reference signal generator (square wave generator) 2, the output of a modulator 3 is connected to one input of an adder 4, and the output of an adder 4 is connected to the output line 16.

[0011] The input-signal line 17 is connected to a wave detector 8 through a band pass filter 7 while connecting with the input of an adder 14. The output of a wave detector 8 and the output of the reference signal generator 2 are connected to the input of a phase comparator 10 through a differentiator 9 and a differentiator 6, respectively while connecting with the input of a level comparator 5.

[0012] The output of a level comparator 5 and the output of a phase comparator 10 are connected to the input of the amount presumption machine 11 of echoes, and the output is connected to the gain-adjustment terminal of the ring buffer 12 and amplifier 13. The sending-signal line 15 is connected to the input of the ring buffer 12, the output of the ring buffer 12 is connected to the input of amplifier 13, the output of amplifier 13 is connected to the input of an adder 14 with the input-signal line 17, and the output of this adder 14 is connected to the input line 18.

[0013] The ring buffer 12 is storage which always samples a sending signal T (sound signal) with the sampling period for 125 microseconds, and stores a minute of data (amplitude) by time series one by cyclically for 250ms.

[0014] Next, actuation is explained. A 2.4kHz**50Hz signal is extracted by the band pass filter 1, the output signal is modulated with a modulator 3 with the output signal (square wave in a cycle of 250ms) of the reference signal generator 2 (amplitude modulation), the output signal of a modulator 3 is added with a sending signal T and an adder 4, and a sending signal T (sound signal) is outputted from an output line 16.

[0015] It is received by the input-signal line 17, and a 2.4kHz**50Hz signal is extracted through a band pass filter 7, a reference signal restores to an input signal R with a wave detector 8, signal amplitude is compared by the output signal (square wave in a cycle of 250ms) and level comparator 5 of the reference signal generator 2, and the gain at the time of transmission of a reference signal and reception is outputted. Furthermore, the signal to which it restored with the wave detector 8 is differentiated with a differentiator 9, the output signal of the reference signal generator 2 is differentiated with a differentiator 6, both standup portions are extracted, a phase is compared by the phase comparator 10, and the time

delay from transmission of a reference signal to reception is outputted to the amount presumption machine 11 of echoes.
[0016] The amount presumption machine 11 of echoes presumes the amount of echoes from the output signal (gain) of a level comparator 5, and the output signal (time delay) of a phase comparator 10, and generates a false echo signal (details after-mentioned). That is, the location which picks out predetermined data from the data (data stored by time series) stored in the ring buffer 12 is specified, and the gain of the amplifier 13 which amplifies said data is directed. Amplifier 13 sets up gain according to directions of the amount presumption machine 11 of echoes, and outputs the signal (false echo signal S) according to said data to an adder 14. An adder 14 subtracts false echo signal S from an input signal R, and outputs it to an input line 18.
[0017] Drawing 2 is a flow chart which shows actuation of the amount presumption machine of echoes. The amount presumption machine 11 of echoes possesses CPU and memory (it omits by a diagram), and the program which performs this flow chart is stored in the memory of the amount presumption machine 11 of echoes, and is performed by CPU. It explains according to the flow chart of drawing 2 . The amount presumption machine 11 of echoes reads the output signal of a level comparator 5 and a phase comparator 10 (step ST 1).
[0018] The ejection location of the ring buffer 12 is decided based on the output signal of a phase comparator 10 (step ST 2). The gain of amplifier 13 is determined based on the output signal of a level comparator 5 (step ST 3). The output (output to amplifier 13) of data is directed to the ring buffer 12 (step ST 4), a setup of gain is directed to amplifier 13, and data is made to output as false echo signal S (step ST 5). The above step ST 1 - step ST 5 are repeated and performed.
[0019] The above-mentioned explanation was Eko-can SENRA by the side of call origination, and the band pass filter 1 extracted the signal of a 2.4kHz**50Hz narrow-band which seldom influences perception in a sending signal T (sound signal), and used with the reference signal (square wave in a cycle of 250ms), having applied the modulation. Moreover, the frequency band of Eko-can SENRA by the side of a call in should just use 2.6kHz**50Hz, in order to prevent the interference by the side of call origination. The frequency and the frequency band on which the above-mentioned reference signal is put, and the period (250ms) of a reference signal and the ring buffer size (a part for storing for 250ms) relevant to it are the values used as an example, and may use other values.
[0020] Since according to this example the amount of echoes is presumed by the time delay from the gain at the time of transmission of a reference signal, and reception, and transmission of a reference signal to reception, false echo-signal S is computed from the data of the stored sound signal and it subtracts from an input signal R as explanation was given [above-mentioned], regardless of echo time, the throughput of the amount presumption machine 11 of echoes becomes fixed, and the message stabilized regardless of the merits and demerits of the echo time by the telephone line can be expected. Moreover, since the subcarrier which sends a reference signal is extracted from a sending signal, its failures, such as a noise, also decrease.
[0021]
[Effect of the Invention] As explained above, according to this invention, the following outstanding effects are expected. Since an echo is presumed by the time delay from the gain at the time of transmission of a reference signal, and reception, and transmission of a reference signal to reception, and the false echo S is computed from the data of the stored sound signal and it subtracts from the received sound signal, the throughput of a false echo output means becomes fixed regardless of the merits and demerits of echo time. Therefore, the message stabilized regardless of the merits and demerits of echo time can be expected, without increasing a hardware element like before. Moreover, since the subcarrier which sends a reference signal is extracted from a sending signal, its failures, such as a noise, also decrease.

TECHNICAL FIELD

[Industrial Application] This invention relates to the echo canceller which negates the echo which returns from a distant office through the telephone line.

PRIOR ART

[Description of the Prior Art] By the long-distance telephone circuit, since its voice is delayed through the telephone equipment of the other party and it returns, it may become an echo and a remarkable failure may be done to a message. The equipment for preventing this is an echo canceller, and is installed in the transceiver circuit of the main unit of telephone equipment etc. Drawing 3 is the block diagram showing the configuration of the conventional echo canceller. The conventional echo canceller possesses the FIR filter (non-recursive filter) 30, the renewal section 34 of a coefficient, and an adder 35 so that it may illustrate. the delay element 31-1 of plurality [filter / 30 / (non-recursive filter) / FIR], 31-2, ..., amplifier 32-1, 32-2, and ... it has an adder 33, and it is constituted so that the coefficient (gain) of the FIR filter 30 may be set up in the renewal section 34 of a coefficient.
[0003] As for each delay element 31-1 of the FIR filter 30, 31-2, and ..., a time delay is set up, it connects with a serial from the sending-signal line 36, amplifier 32-1, 32-2, and ... are connected from each node, and the output of each amplifier is connected to the input of an adder 33. Furthermore, the output of an adder 33 is connected to the input of an adder 35 with the input-signal line 37. The output of an adder 35 is connected to the input of the renewal section 34 of a coefficient, and the output of the renewal section 34 of a coefficient is connected so that the gain of each amplifier may be adjusted.
[0004] While a sending signal is transmitted to a station from the sending-signal line 36, it is inputted into each amplifier through each delay element 31-1, 31-2, and ..., and the output signal of each amplifier is added with an adder 33, and generates false echo signal s. It is inputted into the renewal section 34 of a coefficient while reception of the input signal which received by the input-signal line 37 is carried out, after false echo signal s is subtracted with an adder 35. The renewal section 34 of a coefficient possesses CPU (central processing unit), it adjusts the gain of each amplifier so that the echo signal included in an input signal may become min, and it outputs false echo signal s with the optimal adder 33. The echo signal included in an input signal changes with telephone lines, and optimization of false echo signal s is in drawing by learning the renewal section 34 of a coefficient repeatedly for every telephone line.

EFFECT OF THE INVENTION

[Effect of the Invention] As explained above, according to this invention, the following outstanding effects are expected. Since an echo is presumed by the time delay from the gain at the time of transmission of a reference signal, and reception, and transmission of a reference signal to reception, and the false echo S is computed from the data of the stored sound signal and it subtracts from the received sound signal, the throughput of a false echo output means becomes fixed regardless of the merits and demerits of echo time. Therefore, the message stabilized regardless of the merits and demerits of echo time can be expected, without increasing a hardware element like before. Moreover, since the subcarrier which sends a reference signal is extracted from a sending signal, its failures, such as a noise, also decrease.

TECHNICAL PROBLEM

[Problem(s) to be Solved by the Invention] However, when echo time (time amount until the echo of a sending signal returns) was long, the conventional echo canceller described above needed to increase the delay element of the FIR filter (non-recursive filter) 30, and the number of amplifier, and the processing which optimizes the coefficient (gain) of each amplifier became a huge amount, and it had the problem which says that processing becomes impossible in the renewal section 34 of a coefficient.
[0006] It aims at offering the echo canceller to which this invention was made in view of the above-mentioned point, removes the above-mentioned trouble, and is not concerned with the merits and demerits of echo time, but cancellation throughput becomes fixed.

MEANS

[Means for Solving the Problem] It is the echo canceller which negates an echo signal which this invention is prepared in a telephone set in order to solve the above-mentioned technical problem, and is included in a sound signal from a distant office. A voice data storing means to always sample a sound signal transmitted from a local station, and to store in time series one by one cyclically (12), A reference signal sending-out means to send out a reference signal which extracts a specific frequency component of a sound signal which transmits to a distant office, uses as a subcarrier, and is generated periodically (2), A magnitude-comparison means to compare amplitude of a this reference signal to send out and a reference signal which returns from a distant office, and to ask for a gain (5), A phase-comparison means to find a time delay of a this reference signal to send out and a reference signal which returns from a distant office (10), A false echo output means to adjust this data according to a gain in quest of a predetermined data location of a voice data storing means (12) from this time delay, and to output the false echo S (the amount presumption machine 11 of echoes, and amplifier 13), It is characterized by establishing a means (14) to negate an echo which subtracts this false echo S from a sound signal received from a distant office, and is included in a this received sound signal.

OPERATION

[Function] In this invention, since an echo is presumed by the time delay from the gain at the time of transmission of a reference signal, and reception, and transmission of a reference signal to reception, and the false echo S is computed from the data of the stored sound signal and it subtracts from the received sound signal as explained above, the throughput of a false echo output means becomes fixed regardless of the merits and demerits of echo time. Therefore, the message stabilized regardless of the merits and demerits of echo time can be expected, without increasing a hardware element like before. Moreover, since the subcarrier which sends a reference signal is extracted from a sending signal, its failures, such as a noise, also decrease.

EXAMPLE

[Example] Hereafter, one example of this invention is explained to details based on a drawing. Drawing 1 is the block diagram showing the configuration of the echo canceller of this invention. The echo canceller of this invention possesses a band pass filter 1, the reference signal generator 2, a modulator 3, an adder 4, the level comparator 5, a differentiator 6, a band pass filter 7, a wave detector 8, a differentiator 9, a phase comparator 10, the amount presumption machine 11 of echoes, the ring buffer 12, amplifier 13, and an adder 14 so that it may illustrate.
[0010] The sending-signal line 15 is connected to the input of a band pass filter 1, and one input of an adder 4, the output of a band pass filter 1 is connected to the input of a modulator 3 with the output of the reference signal generator (square wave generator) 2, the output of a modulator 3 is connected to one input of an adder 4, and the output of an adder 4 is connected to the output line 16.
[0011] The input-signal line 17 is connected to a wave detector 8 through a band pass filter 7 while connecting with the input of an adder 14. The output of a wave detector 8 and the output of the reference signal generator 2 are connected to the input of a phase comparator 10 through a differentiator 9 and a differentiator 6, respectively while connecting with the input of a level comparator 5.
[0012] The output of a level comparator 5 and the output of a phase comparator 10 are connected to the input of the amount presumption machine 11 of echoes, and the output is connected to the gain-adjustment terminal of the ring buffer 12 and amplifier 13. The sending-signal line 15 is connected to the input of the ring buffer 12, the output of the ring buffer 12 is connected to the input of amplifier 13, the output of amplifier 13 is connected to the input of an adder 14 with the input-signal line 17, and the output of this adder 14 is connected to the input line 18.
[0013] The ring buffer 12 is storage which always samples a sending signal T (sound signal) with the sampling period for 125 microseconds, and stores a minute of data (amplitude) by time series one by cyclically for 250ms.
[0014] Next, actuation is explained. A 2.4kHz**50Hz signal is extracted by the band pass filter 1, the output signal is modulated with a modulator 3 with the output signal (square wave in a cycle of 250ms) of the reference signal generator 2 (amplitude modulation), the output signal of a modulator 3 is added with a sending signal T and an adder 4, and a sending signal T (sound signal) is outputted from an output line 16.
[0015] It is received by the input-signal line 17, and a 2.4kHz**50Hz signal is extracted through a band pass filter 7, a reference signal restores to an input signal R with a wave detector 8, signal amplitude is compared by the output signal (square wave in a cycle of 250ms) and level comparator 5 of the reference signal generator 2, and the gain at the time of transmission of a reference signal and reception is outputted. Furthermore, the signal to which it restored with the wave detector 8 is differentiated with a differentiator 9, the output signal of the reference signal generator 2 is differentiated with a differentiator 6, both standup portions are extracted, a phase is compared by the phase comparator 10, and the time delay from transmission of a reference signal to reception is outputted to the amount presumption machine 11 of echoes.
[0016] The amount presumption machine 11 of echoes presumes the amount of echoes from the output signal (gain) of a level comparator 5, and the output signal (time delay) of a phase comparator 10, and generates a false echo signal (details after-mentioned). That is, the location which picks out predetermined data from the data (data stored by time series) stored in the ring buffer 12 is specified, and the gain of the amplifier 13 which amplifies said data is directed. Amplifier 13 sets up gain according to directions of the amount presumption machine 11 of echoes, and outputs the signal (false echo signal S) according to said data to an adder 14. An adder 14 subtracts false echo signal S from an input signal R, and outputs it to an input line 18.
[0017] Drawing 2 is a flow chart which shows actuation of the amount presumption machine of echoes. The amount presumption machine 11 of echoes possesses CPU and memory (it omits by a diagram), and the program which performs this flow chart is stored in the memory of the amount presumption machine 11 of echoes, and is performed by CPU. It explains according to the flow chart of drawing 2 . The

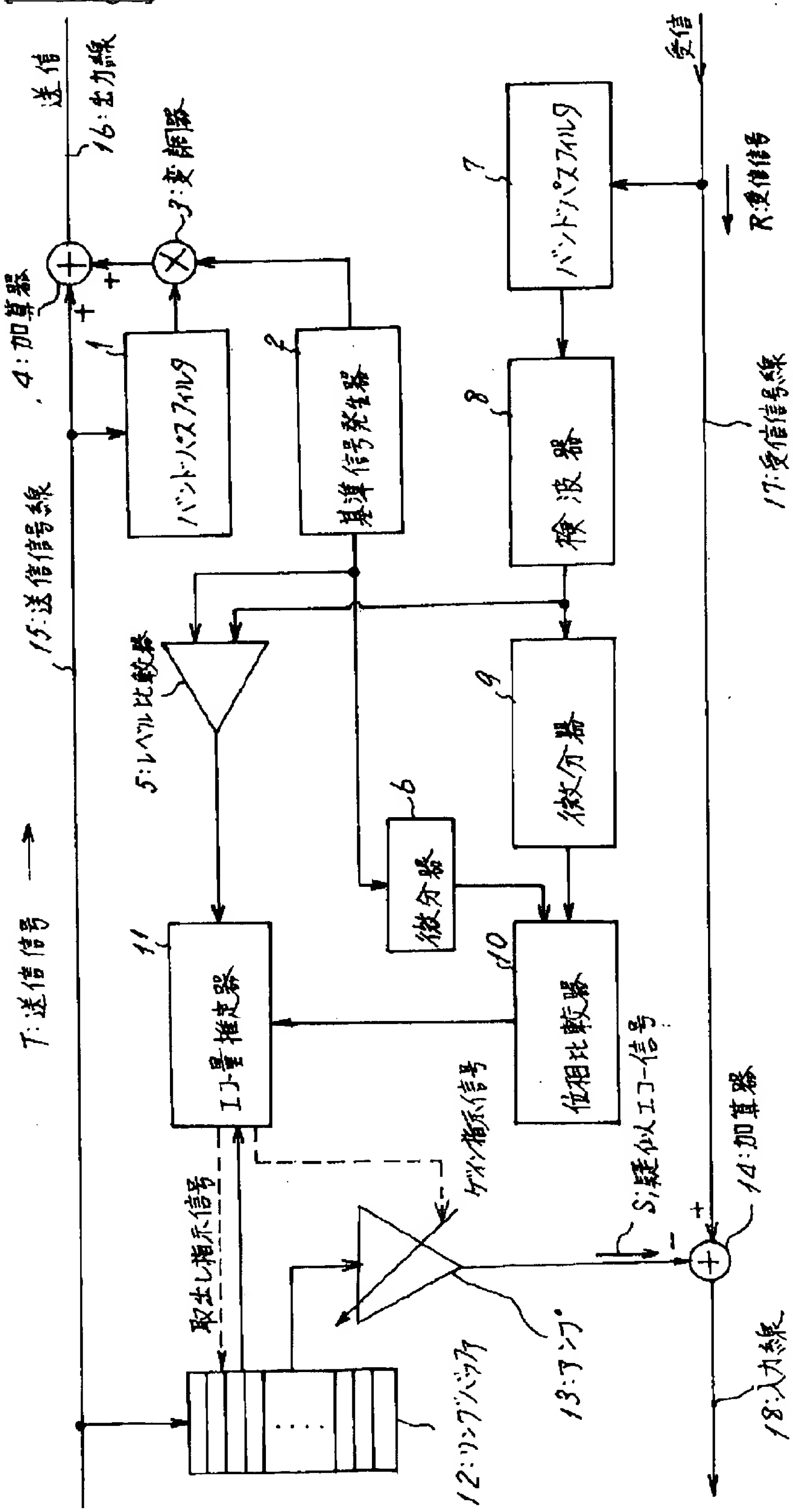
amount presumption machine 11 of echoes reads the output signal of a level comparator 5 and a phase comparator 10 (step ST 1).
[0018] The ejection location of the ring buffer 12 is decided based on the output signal of a phase comparator 10 (step ST 2). The gain of amplifier 13 is determined based on the output signal of a level comparator 5 (step ST 3). The output (output to amplifier 13) of data is directed to the ring buffer 12 (step ST 4), a setup of gain is directed to amplifier 13, and data is made to output as false echo signal S (step ST 5). The above step ST 1 - step ST 5 are repeated and performed.
[0019] The above-mentioned explanation was Eko-can SENRA by the side of call origination, and the band pass filter 1 extracted the signal of a 2.4kHz**50Hz narrow-band which seldom influences perception in a sending signal T (sound signal), and used with the reference signal (square wave in a cycle of 250ms), having applied the modulation. Moreover, the frequency band of Eko-can SENRA by the side of a call in should just use 2.6kHz**50Hz, in order to prevent the interference by the side of call origination. The frequency and the frequency band on which the above-mentioned reference signal is put, and the period (250ms) of a reference signal and the ring buffer size (a part for storing for 250ms) relevant to it are the values used as an example, and may use other values.
[0020] Since according to this example the amount of echoes is presumed by the time delay from the gain at the time of transmission of a reference signal, and reception, and transmission of a reference signal to reception, false echo-signal S is computed from the data of the stored sound signal and it subtracts from an input signal R as explanation was given [above-mentioned], regardless of echo time, the throughput of the amount presumption machine 11 of echoes becomes fixed, and the message stabilized regardless of the merits and demerits of the echo time by the telephone line can be expected. Moreover, since the subcarrier which sends a reference signal is extracted from a sending signal, its failures, such as a noise, also decrease.

DESCRIPTION OF DRAWINGS

[Brief Description of the Drawings]
[Drawing 1] It is the block diagram showing the example of a configuration of the echo canceller of this invention.
[Drawing 2] It is the flow chart which shows actuation of the amount presumption machine of echoes.
[Drawing 3] It is the block diagram showing the example of a configuration of the conventional echo canceller.
[Description of Notations]
1 Band Pass Filter
2 Reference Signal Generator
3 Modulator
4 Adder
5 Level Comparator
6 Differentiator
7 Band Pass Filter
8 Wave Detector
9 Differentiator
10 Phase Comparator
11 The Amount Presumption Machine of Echoes
12 Ring Buffer
13 Amplifier
14 Adder

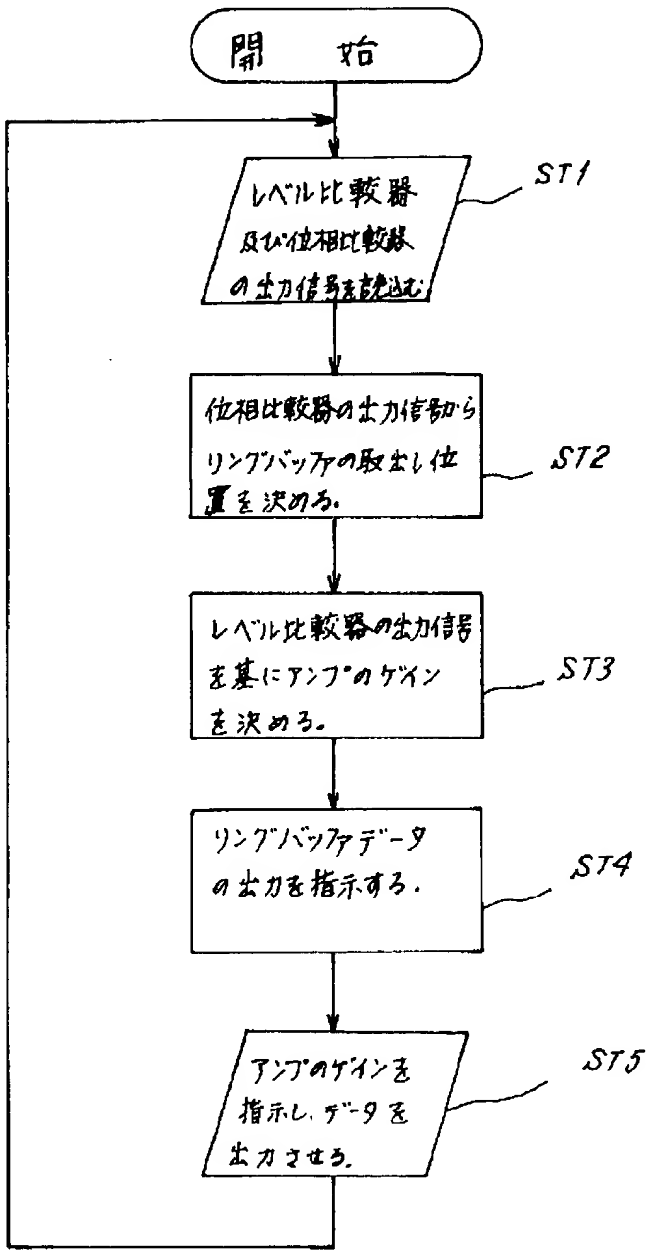
DRAWINGS

[Drawing 1]



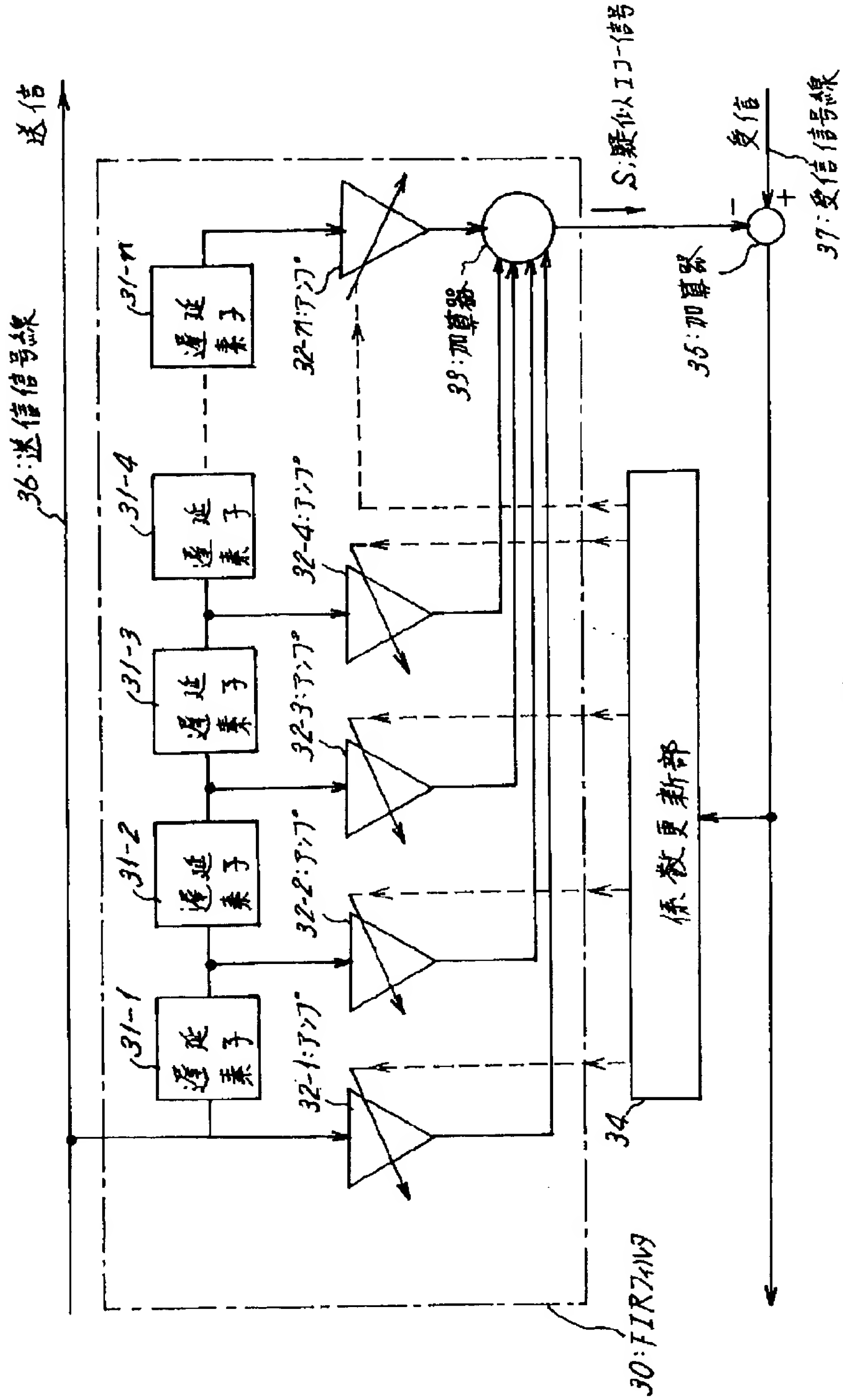
本発明のエコーキャンセラの構成例

[Drawing 2]



エコー量推定器の動作を示すフローチャート

[Drawing 3]



従来のエコーキャンセラのブロック構成例